# Optimal call admission control for voice and data traffic in mobile communication networks

# INSTITUTE OF POLICY AND PLANNING SCIENCES

## Discussion Paper Series

**No. 901**

### Optimal Call Admission Control for Voice and Data Traffic in Mobile Communication Networks

by

Minoru Ohmikawa and Hideaki Takagi

February 2001

UNIVERSITY OF TSUKUBA
Tsukuba, Ibaraki 305-8573
JAPAN

# Optimal Call Admission Control for Voice and Data Traffic in Mobile Communication Networks

Minoru Ohmikawa
Doctoral Program in Policy and Planning Sciences
University of Tsukuba
Tsukuba-shi, Ibaraki 305-8573, Japan
E-mail: mohmikaw@shako.sk.tsukuba.ac.jp

Hideaki Takagi
Institute of Policy and Planning Sciences
University of Tsukuba
Tsukuba-shi, Ibaraki 305-8573, Japan
E-mail: takagi@shako.sk.tsukuba.ac.jp

**Abstract**

In this paper, we propose a new call admission control (CAC) scheme for voice calls in mobile communication networks. Rejection of a hand-off call is more undesirable than that of a new call, for a hand-off call loss causes a severer mental pain. By quantifying their respective pains, we consider them as different costs. The key point of our CAC is restricting new calls in order to minimize the total expected costs per unit time over the long term. An optimal policy is derived from a semi-Markov decision process in which the interval between two consecutive decision epochs is exponentially distributed. Based on this optimal policy, we calculate the steady state probability with respect to the number of established voice connections on the uplink. Thereafter, we evaluate the blocking and forced termination probabilities for voice traffic.

Furthermore, we present an analytic model for the integrated voice and data traffic on the downlink. The state of this model stands for the number of voice calls in OFF state and data packets in the system at the beginning of every frame on condition that the number of voice connections is fixed. After the steady state probability for this model is calculated, we examine the effect of CAC on the throughput and waiting time of data packets at low, middle and high voice load conditions.

We find that the forced termination probability is improved by our CAC scheme at the sacrifice of the blocking probability and channel utilization for voice calls. Besides our CAC is effective for data traffic at middle and high voice load condition.

**Keywords:** call level, frame level, call admission control, semi-Markov decision process, CDMA, QoS, packet error probability

# 1 Introduction

Recent years have witnessed a remarkable development in mobile communication techniques. The population of mobile users is explosively growing with the technological progress. Application services provided today are diverse, including cellular telephony, e-mail and the Internet browsing. It is predicted that data traffic will occupy high proportion of entire traffic in the near future, so the call admission control (CAC) becomes a mandatory element of mobile communication networks.

A direct-sequence code division multiple access (DS-CDMA) scheme gets much attention as one of the third generation access protocols, for it improves not only channel capacity but also security and interference. In order to guarantee in advance the specified quality of service (QoS) requirements, the number of users that can be simultaneously accommodated must be regulated, because CDMA realizes multiple access by a wideband averaging technique which reduces the interference by averaging it over a long time interval.

Many researchers have tackled with approximations of calculating the bit error probability (BEP) in the DS-CDMA system. In [1], expressions are developed for the multiple access interference (MAI) and the upper and lower bounds are given on the average error probability for a direct-sequence spread spectrum multiple access (DS-SSMA) system. An accurate approximation, called "improved Gaussian approximation," is subsequently presented in [2]. Another approximation that is simpler but maintains the same accuracy is proposed in [3], which is further simplified in [4].

As for the CAC, many schemes have been proposed; however, few have taken hand-off calls into account. A CAC scheme is implemented in [5] by employing a semi-Markov decision process. But hand-off calls are not considered in this model. Besides the voice activity factor is used instead of an on-off model for each call, because it is assumed that the CDMA system monitoring cannot draw the line between talkspurt and silence periods for each call. In [6] the congestion control is carried out for data traffic to ensure that the packet error probability (PEP) of voice be lower than a specified QoS requirement. To fully utilize the channel capacity, the congestion control dynamically updates the probability that users who are assigned special codes start transmitting their data packets.

In this paper, we consider a CAC scheme by including hand-off calls. We investigate to what degree it has an impact on the performance of mobile communication networks. Our system is modeled in two time scales, that is, a call level (the order of a few minutes) and a frame level (the order of a few ten milliseconds). A CAC scheme dealing with hand-off calls is executed in the call level. Once the best policy is obtained as the result of CAC, the hand-off rate is computed from the probabilistic analysis of the behavior of mobile terminals as in [7]–[10]. The steady state probability in the call level model is used to calculate the blocking and forced termination probabilities for voice traffic. In addition a frame level model which meets a QoS requirement for data is proposed. We consider an on-off model for each admitted call and a two-dimensional Markov chain for integrated voice and data traffic. The steady state probability in the frame level model is used to calculate the packet throughput and waiting time for data traffic.

The rest of this paper is organized as follows. Section 2 describes our call and frame level models. Section 3 defines performance measures for voice and data traffic. Section 4 presents numerical results. Conclusions are drawn in Section 5.

# 2 System Model

Let us distinguish the time scale of data traffic from that of voice traffic. Data packets transmitted by mobile users are first grouped into frames the length of which is a few tens of milliseconds, then encoded, repeated to adjust the data rate, and interleaved. On the other hand, voice calls have call holding and cell residence times of which the length is a few minutes. If the system is assumed to be stable in the frame level before the number of calls changes in the call level, then it is reasonable that the system is independently modeled in two time scales. Hence, we consider a call level model in which the time unit is one minute and a frame level model in which the time unit, which equals to the length of a frame, is 20 milliseconds.

Voice has priority over data due to its delay sensitive property. Channels are first assigned to all the voice users and the leftover is used for data users. We exclude the case in which the transmission of data traffic causes the system to reject new or hand-off voice calls. Thus we can treat only the number of voice connections in the call level. On the assumption that it is fixed, we deal with integrated voice and data traffic in the frame level. In the following subsections, we define these models specifically, and analyze them.

## 2.1 Call level model

In this subsection, we propose a call level model for a cell in a cellular communication network. We follow a chain of events a call experiences in the cell. We present a CAC method using a semi-Markov decision process.

### 2.1.1 Imbedded Markov chain

Let us pay our attention to a single cell. The cell can accommodate calls up to the multiple access capability (MAC) for voice users $K^v$, which is determined by a specified QoS requirement. If a new call is placed and the number of existing calls is less than $K^v$, then the call is admitted into the cell; otherwise, the call is blocked. Similarly, if a hand-off call arrives and the number of existing calls is less than $K^v$, then the hand-off process succeeds; otherwise, the call is forcibly terminated (i.e. the hand-off process fails).

The average call holding and cell residence times are denoted by $1/\mu_1$ and $1/\mu_2$, respectively. Both times are assumed to be exponentially distributed. Let $\lambda_1$ and $\lambda_2$ be the rates at which new and hand-off calls are generated respectively according to independent Poisson processes.

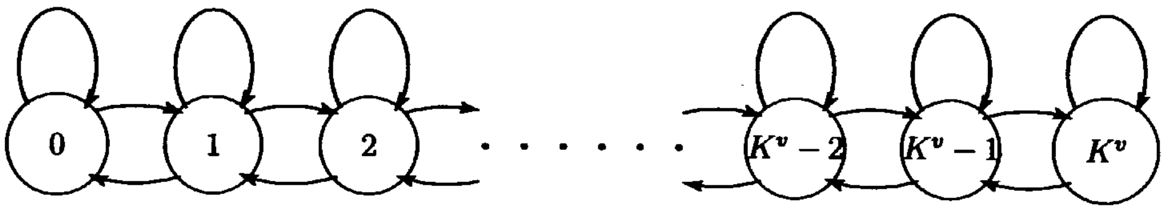


Figure 1: State transition diagram.

Consider an imbedded Markov chain [11] in which the state is defined by the number of calls simultaneously existing in the cell. In this method, we identify a set of decision epochs such that, if we specify the state in the call level model and also provide information on the decision epoch thereafter, then we can know the state in the call level model at the next such decision epoch. We will take up these decision epochs in detail in the next subsection. There are three cases with respect to the one step transition from state $i$. First, if a new or hand-off call is placed in a cell and is admitted into the cell, then the chain moves from state $i$ to state $i+1$. Second, if the call is placed but rejected, then the chain stays in state $i$. Third, if the call is completed or leaves the cell as a hand-off call, then the chain moves from state $i$ to state $i-1$. Figure 1 depicts these transitions. These transition probabilities are denoted by $q_{i,i+1}, q_{i,i}$ and $q_{i,i-1}; i = 0, 1, \cdots, K^v$, respectively. They are given in terms of $\mu_1, \mu_2, \lambda_1, \lambda_2$, and the parameter of CAC; see equations (9)–(11) below. Since $q_{i,j} = 0$ if $|i-j| > 1$, the Markov chain is a birth-and-death process. Let $p_j; j = 0, 1, \cdots, K^v$ be the steady state probability for the chain to be found in state $j$. Then we can have the set of linear equations

$$\begin{cases} p_j = \sum_{i=0}^{K^v} p_i q_{i,j} \qquad j = 0, 1, \cdots, K^v \\ \sum_{j=0}^{K^v} p_j = 1 \end{cases} \tag{1}$$

which has a positive solution.

Given the steady state probabilities for the number of simultaneous calls in the cell, the inbound hand-off rate from neighboring cells $\lambda_2$ can be determined by the fixed-point method [9]. Suppose that the network consists of independent and statistically identical cells, the rate $\lambda_2$ can be considered to be equal to the outbound hand-off rate to adjacent cells $\lambda_2^*$. The algorithm to calculate the rate $\lambda_2$ is as follows.

1. Initialize $\lambda_2 = \lambda_2^* = 0$.
2. Calculate $p_j; j = 0, 1, \cdots, K^v$ by solving equations (1).
3. Compute
$$\lambda_2 = \lambda_2^* = \sum_{j=1}^{K^v} p_j j \mu_2. \tag{2}$$
4. Repeat steps **2** and **3** until $\lambda_2$ converges.

### 2.1.2 Call admission control policy

Accidents that ongoing conversations are forcibly terminated generally displease us more than initial access failures. Thus it makes sense that we weight rejection costs of new and hand-off calls and minimize the average cost per unit time over the long term. The cost should quantify the strength of stress which rejected customers feel.

A policy which rejects a new call even if there are available channels may reject fewer hand-off calls on the average than another policy which accepts a new call whenever there are available channels. Here, the word "policy" means a compass which indicates an action to take in all state. Finding the optimal policy is the goal of our CAC. In other words, the optimal choice of actions (accept or reject) is made in each state to minimize the average cost per unit time over the long term.

In order to find the optimal policy, we employ a semi-Markov decision process as in [5]. The process is observed when a conversation is completed, a calling user goes out of the cell, a new call is placed, or a hand-off call arrives. Since each of these events occurs in a Poisson process, the interval between two consecutive observation points, called decision epochs, is exponentially distributed. After observing the process, a decision has to be made according to the policy and the corresponding cost is incurred as a consequence of the decision made. A set of possible states is defined by

$$I = \{\mathbf{x} = (i,j) | i = 0, 1, \cdots, K^v; j = 0, 1, 2\} \tag{3}$$

where $i$ is the number of calls simultaneously existing in the cell; $j$ represents the kind of decision epochs. When $j = 0$, which means that a conversation completion or a caller going out of the cell is observed, the epoch is referred to as a fictitious decision epoch because no cost is incurred on account of no call rejection. When $j = 1$ or 2, where the former means that a new call arrival is observed and the latter means that a hand-off call arrival is observed, these epochs are referred to as real decision epochs and a desirable choice $a \in \{0, 1\}$ must be made. Here, the rejection and acceptance are denoted by 0 and 1, respectively.

Suppose that the cost function is given by

$$C_{\mathbf{x}}(a) = (1-a)\gamma_j \qquad a \in \{0, 1\} \tag{4}$$

where $C_{\mathbf{x}}(a)$ indicates the expected cost incurred until the next decision epoch if action $a$ is chosen in state $\mathbf{x} = (i, j)$; $\gamma_j$ quantifies the strength of stress which a rejected user feels $j = 0, 1, 2$, where $\gamma_0 = 0$.

Let the expected time until the next decision epoch when action $a$ is chosen in state $\mathbf{x}$ be denoted by $\tau_{\mathbf{x}}(a)$. Then the following equations hold due to the memoryless property at decision epochs.

$$\tau_{\mathbf{x}}(a) = \begin{cases} \dfrac{1}{\lambda_1 + \lambda_2 + (i-1)(\mu_1 + \mu_2)} & \mathbf{x} = (i, 0) \\ \dfrac{a}{\lambda_1 + \lambda_2 + (i+1)(\mu_1 + \mu_2)} + \dfrac{1-a}{\lambda_1 + \lambda_2 + i(\mu_1 + \mu_2)} & \mathbf{x} = (i, 1 \text{ or } 2) \end{cases} \tag{5}$$

According to the memoryless property, the following state transition probabilities are derived.

$$P_{\mathbf{x},\mathbf{x}'}(a) = \begin{cases} \left.\begin{array}{ll} \dfrac{\lambda_1}{\lambda_1 + \lambda_2 + (i-1)(\mu_1 + \mu_2)} & \mathbf{x}' = (i-1, 1) \\ \dfrac{\lambda_2}{\lambda_1 + \lambda_2 + (i-1)(\mu_1 + \mu_2)} & \mathbf{x}' = (i-1, 2) \\ \dfrac{(i-1)(\mu_1 + \mu_2)}{\lambda_1 + \lambda_2 + (i-1)(\mu_1 + \mu_2)} & \mathbf{x}' = (i-1, 0) \end{array}\right\} & \mathbf{x} = (i, 0) \\ \left.\begin{array}{ll} \dfrac{\lambda_1}{\lambda_1 + \lambda_2 + (i+a)(\mu_1 + \mu_2)} & \mathbf{x}' = (i+a, 1) \\ \dfrac{\lambda_2}{\lambda_1 + \lambda_2 + (i+a)(\mu_1 + \mu_2)} & \mathbf{x}' = (i+a, 2) \\ \dfrac{(i+a)(\mu_1 + \mu_2)}{\lambda_1 + \lambda_2 + (i+a)(\mu_1 + \mu_2)} & \mathbf{x}' = (i+a, 0) \end{array}\right\} & \mathbf{x} = (i, 1 \text{ or } 2) \end{cases} \tag{6}$$

where $\mathbf{x}'$ denotes the next state.

Now, the value-iteration algorithm of a semi-Markov decision process [12] is applied to our CAC model in order to determine the optimal policy. To do so, we need to convert a semi-Markov decision model into a discrete-time Markov decision model such that the average costs per unit time over the long term of each stationary policy are the same in both models.

If the current state is $\mathbf{x}$ and action $a$ is chosen, then the average cost per unit time until the next decision epoch is given by $C_{\mathbf{x}}(a)/\tau_{\mathbf{x}}(a)$. If a constant $\tau$ is chosen to satisfy the condition $0 < \tau \leq \min_{\mathbf{x},a} \tau_{\mathbf{x}}(a)$, then a decision epoch occurs in time $\tau$ with probability $\tau/\tau_{\mathbf{x}}(a)$. The quantity $V_n(\mathbf{x})$ is introduced as the minimal total expected cost with $n$ decision epochs left to the time horizon when the current state is $\mathbf{x}$. A terminal cost $V_0(\mathbf{x}'')$ is incurred if the process ends at state $\mathbf{x}''$. Since the key point is minimizing the average cost per unit time over the long term, we must go backward in the time axis until the cost $V_n(\cdot) - V_{n-1}(\cdot)$ per unit time converges. This circumstance is illustrated in Figure 2.

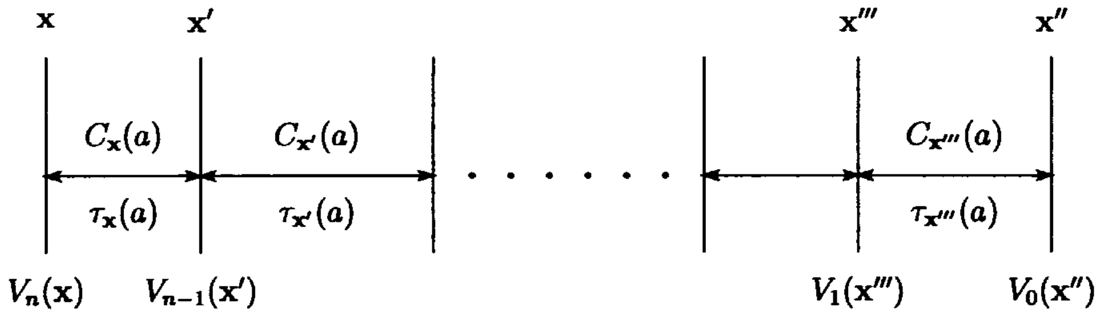


Figure 2: Diagram for the value-iteration algorithm.

Considering the original semi-Markov decision model as a discrete-time Markov decision model, the value-iteration algorithm to find the optimal policy can be written as follows.

1. Choose $V_0(\mathbf{x})$ such that $0 \leq V_0(\mathbf{x}) \leq \min_a\{C_{\mathbf{x}}(a)/\tau_{\mathbf{x}}(a)\}$ for all $\mathbf{x}$. Set $n = 1$.
2. Compute for all $\mathbf{x}$

$$V_n(\mathbf{x}) = \min_{a\in\{0,1\}} \left[\frac{C_{\mathbf{x}}(a)}{\tau_{\mathbf{x}}(a)} + \frac{\tau}{\tau_{\mathbf{x}}(a)} \sum_{\mathbf{x}'\in I} P_{\mathbf{x},\mathbf{x}'}(a)V_{n-1}(\mathbf{x}') + \left\{1 - \frac{\tau}{\tau_{\mathbf{x}}(a)}\right\} V_{n-1}(\mathbf{x})\right]. \quad (7)$$

3. Compute the upper and lower bounds

$$M_n = \max_{\mathbf{x}\in I}\{V_n(\mathbf{x}) - V_{n-1}(\mathbf{x})\} \quad \text{and} \quad m_n = \min_{\mathbf{x}\in I}\{V_n(\mathbf{x}) - V_{n-1}(\mathbf{x})\}. \quad (8)$$

4. If $0 \leq (M_n - m_n) \leq \varepsilon m_n$, then the algorithm is stopped with the optimal policy. Otherwise $n = n + 1$ and go to step 2.

Here, $\varepsilon$ is a small positive constant for stopping the iteration. In our numerical experiments, the algorithm had convergence with $\varepsilon = 10^{-3}$.

It is proved in [13] that this algorithm will be stopped after finitely many iterations under the assumption that the associated Markov chain is aperiodic for each stationary policy. However, even though the Markov chain is periodic, the periodicity can be circumvented by a perturbation of the one step transition probabilities [12]. Furthermore, it is shown in [14] the value iteration method exhibits a geometric rate of convergence, whenever convergence occurs.

Once the optimal policy $\{\hat{a}_{(i,j)} | i = 0, 1, \cdots, K^v; j = 0, 1, 2\}$ is determined, the state transition probabilities $q_{i,i+1}, q_{i,i}$ and $q_{i,i-1}$ in equation (1) are derived. Namely,

$$\begin{aligned} q_{i,i+1} &= \sum_{j=0}^{2} \hat{a}_{(i,j)} P(j|i) \\ &= \frac{\hat{a}_{(i,1)}\lambda_1 + \hat{a}_{(i,2)}\lambda_2}{\lambda_1 + \lambda_2 + i(\mu_1 + \mu_2)} \end{aligned} \tag{9}$$

where $\hat{a}_{(i,0)} = 0$; $P(j|i)$ is the conditional probability that the kind of the next decision epoch is $j$ on the condition that there are $i$ calls in the cell. Similarly, we get

$$q_{i,i} = \frac{(1 - \hat{a}_{(i,1)})\lambda_1 + (1 - \hat{a}_{(i,2)})\lambda_2}{\lambda_1 + \lambda_2 + i(\mu_1 + \mu_2)} \tag{10}$$

$$q_{i,i-1} = \frac{i(\mu_1 + \mu_2)}{\lambda_1 + \lambda_2 + i(\mu_1 + \mu_2)}. \tag{11}$$

Finally, we substitute these transition probabilities into equation (1) to obtain the steady state probabilities $p_j; j = 0, 1, \cdots, K^v$ for a cell under the optimal CAC policy.

## 2.2 Frame level model

In this section, another model is considered concerning the integrated voice and data traffic on the downlink (from base station to mobile terminals) in the frame level on the condition that the number of established voice connections (i.e. admitted calls) is fixed.

### 2.2.1 Talkspurt and silence periods

As each voice user admitted into the system alternates talkspurt and silence periods, every call is assumed to be in either "ON" or "OFF" state. If a call is in ON state, then the user is speaking (i.e. his/her voice traffic is transmitted on the uplink). If a call is in OFF state, then the user is listening to the other party's voice (i.e. his/her voice traffic is transmitted on the downlink) or both end users are silent (i.e. no voice traffic is transmitted). However, the last case is assumed to be rare, and it is ignored. Thus, for the traffic on the downlink, what is important is the number of calls in OFF state, not that in ON state.

Let the average lengths of talkspurt and silence periods be denoted by $\xi$ and $\eta$, respectively. If these periods are assumed to be distributed exponentially, then the state transition probabilities $\alpha$ and $\beta$ from ON to OFF and OFF to ON in a frame of length $T$ can be calculated respectively as

$$\alpha = 1 - \exp(-\frac{T}{\xi}) \tag{12}$$

$$\beta = 1 - \exp(-\frac{T}{\eta}). \tag{13}$$

This two-state Markov chain is shown in Figure 3.

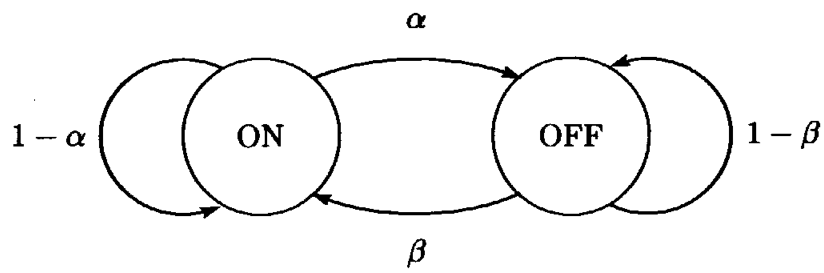


Figure 3: ON-OFF state transition diagram for each user.

If there are $m$ established voice connections at any instant of time, then their transitions between ON and OFF are independent of each other, and never affected by data packets because voice calls are assumed to be always served prior to data packets. Thus, the number of transitions between ON and OFF follows a binomial distribution. It is also assumed that the system reaches the steady state in the frame level before the number of calls (i.e. $m$ established voice connections) changes in the call level. Therefore, the following state transition equation holds for the number of calls in the frame level.

$$N_{t+1}^v = N_t^v + B(m - N_t^v, \alpha) - B(N_t^v, \beta) \tag{14}$$

where $N_t^v$ is a random variable which stands for the number of calls in OFF state at the beginning of the $t$th frame; $B(n, \alpha)$ is a binomial random variable with parameters $n$ and $\alpha$.

It is convenient for the numerical calculation to replace equation (14) with the following equation in terms of the state transition probabilities:

$$P^v_{i,i'}(m) = \sum_{l=0}^{m-i} \binom{m-i}{l} \alpha^l (1-\alpha)^{m-i-l} \binom{i}{i+l-i'} \beta^{i+l-i'} (1-\beta)^{i'-l} \qquad 0 \le i, i' \le m. \tag{15}$$

$P^v_{i,i'}(m)$ is the probability that the number of calls in OFF state changes from $i$ to $i'$ in a frame when there are $m$ established voice connections.

### 2.2.2 Markov chain for data traffic process

Since data traffic is more demanding than voice with respect to the PEP as a QoS requirement, the MAC for data users $K^d$ is usually set to be less than that for voice users $K^v$. But data is delay insensitive compared with voice. Thus data packets are transmitted only if there are channels available which can transmit them while meeting their QoS requirements on the downlink; otherwise they must wait until any channel becomes free. Alternatively, the number of channels available for data packets is dependent on the number of calls in OFF state at the beginning of a frame. A channel must serve as long as the data buffer is not empty. It is assumed that exactly one packet per frame is served by one channel.

Consequently, transmitting data packets never interrupts the state transition between ON and OFF of each established voice connection. On the other hand, the QoS requirement for data is never violated by squeezing the state transition of each voice connection, for the transmission of one packet never goes over multiple frames. To understand this frame level model more concretely, the reader is referred to Figure 4.

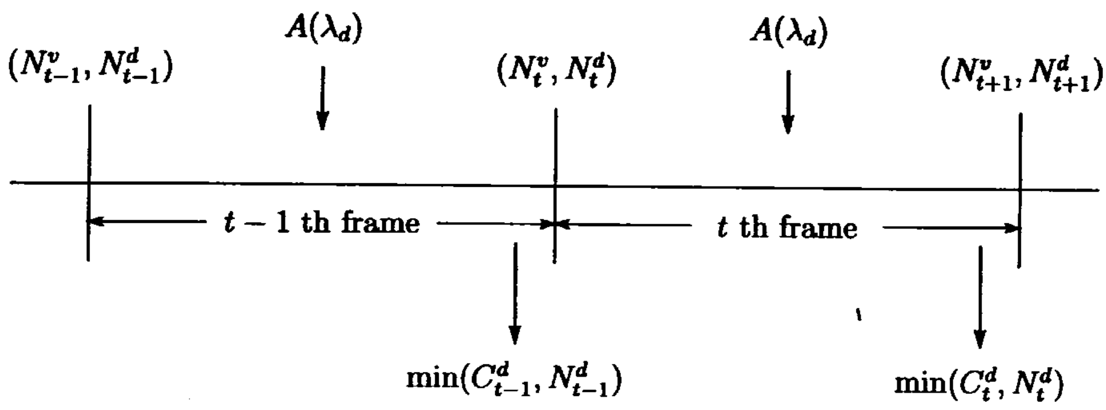


Figure 4: Sketch of the frame level model.

Here $N^d_t$ denotes the number of data packets in the system. The state $(N^v_t, N^d_t)$ means that there are $N^v_t$ calls in OFF state and $N^d_t$ packets in the system at the beginning of the $t$th frame. A Poisson random variable with parameter $\lambda_d$, which is the average number of data packets arriving in a frame, is denoted by $A(\lambda_d)$. The number of channels available at the beginning of the $t$th frame $C_t^d$ is given by

$$C_t^d = \max(K^d - N_t^v, 0) \tag{16}$$

Thus the base station transmits $\min(C_t^d, N_t^d)$ packets in the $t$th frame. Figure 4 can be put into words as follows. Data packet arrivals occur randomly. Those that have arrived by the beginning of a frame may be served in that frame. Packet departures take place just before the end of each frame. Moreover Figure 4 helps to derive the state transition equation at frame boundaries as

$$N_{t+1}^d = N_t^d + A(\lambda_d) - \min(C_t^d, N_t^d). \tag{17}$$

When $m$ calls exist in the system, let $I'(m)$ represent a set of combinations $\mathbf{n} = (i, j)$, where $i$ is the number of calls in OFF state and $j$ is the number of data packets in the system at the beginning of a frame:

$$I'(m) = \{\mathbf{n} = (i, j) | i = 0, 1, \cdots, m; j = 0, 1, \cdots, K^d + K^b\}. \tag{18}$$

Here $K^b$ denote the buffer size for data packets; thus $K^d + K^b$ is the maximum number of data packets in the system, which occurs when there is no voice connection in the cell. We need the steady state probability $\pi_{(i,j)}; (i, j) \in I'(m)$ for the process $(N_t^v, N_t^d)$ that evolves according to equations (14) and (17). This steady state probability will be used to calculate various performance measures as discussed later.

Additionally, replacing the equation (17) with the state transition probability is helpful to conduct the numerical experiment. Suppose that the state changes from $\mathbf{n} = (i, j)$ to $\mathbf{n}' = (i', j')$ and the number of packets increases by $\delta$ in a frame. We then have

$$\delta = \delta(i, j, i', j') := j' - j + \min\{c_d(i), j\} \tag{19}$$

where $c_d(i)$ is the value of $C_t^d$, and $\min\{c_d(i), j\}$ is the number of data packets actually transmitted in a frame when the state is $\mathbf{n} = (i, j)$ at the beginning of the frame. Consequently, the state transition probabilities in the frame level model are obtained as

$$P_{(i,j),(i',j')}^d(m) = \begin{cases} P_{i,i'}^v(m) \dfrac{{\lambda_d}^\delta \exp(-\lambda_d)}{\delta!} & j' < K^b \\ P_{i,i'}^v(m) \displaystyle\sum_{h=\delta}^{\infty} \dfrac{{\lambda_d}^h \exp(-\lambda_d)}{h!} & j' = K^b \\ 0 & j' > K^b \end{cases} \tag{20}$$

where $P_{i,i'}^v(m)$ is given in equation (15).

# 3 Performance Measures

In this section, we introduce some performance measures suitable for each traffic type in order to compare the CAC model with the non-CAC model. Namely, we use the blocking and forced termination probabilities for voice traffic and the packet throughput and waiting time for data traffic.

## 3.1 Multiple access capability

In this paper, the DS-CDMA protocol is considered as a multiple access technique. It is an averaging system which reduces the interference by averaging the signal power over a long time interval. Therefore, the more multiple DS signals overlap in time and frequency the noisier the received signal is, resulting in the degradation of QoS. To guarantee various QoS requirements, we must adjust the MAC for each type of users. We consider only the PEP as a specified QoS requirement.

Various approximation techniques for the BEP have been developed for the DS-CDMA radio system using binary phase shift keyed (BPSK) signaling. In particular, the improved Gaussian approximation in [2] is accurate. In this paper we use an approximation given in [4], which simplifies the improved Gaussian approximation while maintaining the same accuracy.

It is assumed that interfering signal sequences are random and that the MAI is the only source of bit errors. Suppose that the ratio $N = T_b/T_c$ is a constant, where $T_b$ is the duration of each encoded data bit and $T_c$ is the duration of each chip in the signal sequence, and that the system has $K$ simultaneous users. It is well known that the BEP can be approximately calculated by finding the average signal-to-noise ratio (SNR) $x$ and using the $Q$ function

$$Q(x) = \frac{1}{\sqrt{2\pi}} \int_x^\infty \exp(-\frac{u^2}{2}) du. \tag{21}$$

According to [4], the simplified improved Gaussian approximation yields results close to the improved Gaussian approximation as long as the BEP is larger than about $10^{-6}$. If the mean $\mu$ and the variance $\sigma$ of the distribution of the MAI variance are available, then the BEP is given by

$$\mathrm{BEP}(K) = \frac{2}{3}a + \frac{1}{6}b + \frac{1}{6}c \tag{22}$$

where

$$a = Q\left(\sqrt{\frac{N^2}{\mu}}\right) \tag{23}$$

$$b = Q\left(\sqrt{\frac{N^2}{\mu + \sqrt{3}\sigma}}\right) \tag{24}$$

$$c = Q\left(\sqrt{\frac{N^2}{\mu - \sqrt{3}\sigma}}\right). \tag{25}$$

The values of $\mu$ and $\sigma$ are given by

$$\mu = (K-1)\frac{N}{3} \tag{26}$$

$$\sigma \approx (K-1)\frac{23N^2}{360}. \tag{27}$$

Let us now consider the PEP using above the BEP. We can express the PEP in a closed form if we ignore the bit-by-bit error dependency for ease of computation. Suppose that one packet length is $L$ bits and the block error correction capability is incorporated into the data packet which can correct $t$ or fewer bits. Then the PEP becomes as follows.

$$\mathrm{PEP}(K) = 1 - \sum_{i=0}^{t} \binom{L}{i} \mathrm{BEP}(K)^i (1 - \mathrm{BEP}(K))^{L-i} \tag{28}$$

This equation yields two MACs, $K^v$ for voice and $K^d$ for data, if $\mathrm{PEP}(K)$ satisfy their QoS requirements: e.g., $\mathrm{PEP}(K^v) < 10^{-2}$ for voice and $\mathrm{PEP}(K^d) < 10^{-5}$ for data.

## 3.2 Blocking and forced termination probabilities

There are two situations which irritate users. One is that users who newly try to call someone are rejected by the system due to no available channel (i.e. blocking new calls). The other is that users who are on the phone and crossing the cell boundary are forcibly broken off (i.e. forced termination of hand-off calls). We introduce new call blocking and hand-off call forced termination probabilities as performance measures in the call level.

If a new call is placed in a cell and the action which follows the optimal policy is "reject", then the rejection is regarded as blocking of a new call. If a hand-off call arrives in a cell and is rejected according to the optimal policy, then the rejection is regarded as forced termination of a hand-off call. Let $Pb$ and $Pf$ represent the blocking and forced termination probabilities, respectively. Then

$$Pb = \frac{\lambda_1}{\lambda_1+\lambda_2} \sum_{\hat{a}(i,1)=0} p_i \qquad (i,1) \in I \tag{29}$$

$$Pf = \frac{\lambda_2}{\lambda_1+\lambda_2} \sum_{\hat{a}(i,2)=0} p_i \qquad (i,2) \in I \tag{30}$$

where $p_i; i = 0, 1, \cdots, K^v$ is the steady state probability for the number of calls in a cell, given as the solution to equation (1).

### 3.3 Throughput and waiting time

It is pointed out in the previous section that the number of simultaneously transmitted data packets has to be less than $K^d$ for data users in order to guarantee the QoS requirement for data. Since the number of channels available for data packets varies with the number of calls in OFF state, the packet throughput is not constant even when the data buffer is full of packets waiting to be transmitted. When a packet arrives at any instant of time, it must wait till the beginning of the next frame even though there are some idle channels. The waiting time in the data buffer (i.e. queueing delay) depends on the packet throughput.

Consider the steady state $(i, j) \in I'(m)$ on the condition that $m$ voice connections are established, where we have $i$ calls in OFF state and $j$ data packets in the system at the beginning of a frame. If the steady state probability $\pi_{(i,j)}$ is obtained, then the packet throughput $S_d(m)$, conditioned on $m$ established voice connections, can be calculated as follows.

$$S_d(m) = \sum_{(i,j)\in I'(m)} \min\{c_d(i), j\}\{1 - \mathrm{PEP}(i+j)\}\pi_{(i,j)} \tag{31}$$

The average packet throughput $S_d$ is then given by

$$S_d = \sum_{m=0}^{K^v} p_m S_d(m). \tag{32}$$

Next, we focus on the average waiting time $W_d$, the interval between the arrival and departure times of a packet counted in the number of frames. Note that the average number of data packets in the system $N_d(m)$, conditioned on $m$ established voice connections is given by

$$N_d(m) = \sum_{(i,j)\in I'(m)} j\pi_{(i,j)} \tag{33}$$

Thus the average waiting time $W_d$ is calculated by

$$W_d = \frac{1}{2} + \frac{\sum_{m=0}^{K^v} p_m N_d(m)}{\sum_{m=0}^{K^v} p_m S_d(m)} = \frac{1}{2} + \frac{1}{S_d} \sum_{m=0}^{K^v} p_m N_d(m). \tag{34}$$

where 1 is 1/2 is the average time from the arrival time of a packet to the beginning of the next frame because of random arrivals in a frame; the last term in the right-hand side is the average queueing delay by Little's law.

# 4 Numerical Experiments

In this section, numerical experiments are carried out in order to evaluate to what degree our CAC scheme has an impact on the performance of voice and data traffic. It is assumed that the average call holding time $1/\mu_1 = 1.5$ and the average cell residence time $1/\mu_2 = 5$ (minutes); Japanese average talkspurt period $\xi = 0.6$ and the average silence period $\eta = 0.9$ (seconds) [15]; the size of the data buffer $K^b = 20$ (packets).

We begin with determining the values of MACs $K^v$ and $K^d$ for voice and data users, respectively. We examine the effectiveness of the CAC in the call level by means of the blocking and forced termination probabilities for voice calls. We conclude with the assessment of the CAC in the frame level by the throughput and waiting time of data packets.

## 4.1 Results and discussions

The IS-95 air interface [16] says that the signal at a rate of 19.2 Kbps is spread with an orthogonal Walsh code at a rate of 1.2288 Mcps, that is, $N = 1228.8/19.2 = 64$. Data packets to be transmitted are first grouped into 20 milliseconds frames ($T = 20 \times 10^{-3}$). Since it is assumed that exactly one packet is transmitted in a frame and the bit rate is 14.4 Kbps, the packet length $L$ is $14.4 \times 20 = 288$ bits. Furthermore, it is assumed that one packet includes the block error correction capability that can correct two or fewer bit errors ($t = 2$ in equation (28)).

We have discussed in section 3.1 that the MAC is dependent on system parameters and specified QoS requirements. The system has to ensure that the PEP of transmitted data packets never violates the specified QoS requirement. Referring to Figure 5, obtained by equation (28), we can find that $K^v = 22$ and $K^d = 14$, which correspond to such numbers of simultaneous users that the PEP is lower than $10^{-2}$ for voice and $10^{-5}$ for data. From now on, $K^v = 22$ and $K^d = 14$ are fixed when calculating various performance measures.

Figure 6 shows the blocking probability $Pb$ as a function of new call arrival rate $\lambda_1$.

Here, ratio=1 means the absence of CAC in the sense that the system equally works for all calls generated in a cell whether the calls are new or hand-off. Ratio=10 and ratio=100 mean that the rejection of hand-off calls gives users 10 and 100 times more stress than new calls. If the system restricts new calls for hand-off calls, then it is natural that the larger ratio, the larger the blocking probability. The blocking probabilities after implementation of the CAC with ratio=10 and ratio=100 are 2.6 and 5.8 times, respectively, larger than that under the non-CAC at $\lambda_1 = 6$.

In Figure 7, we plot the forced termination probability $Pf$ as a function of new call arrival rate $\lambda_1$. As hand-off calls have great advantage over new calls under the CAC with large ratio, the forced termination probabilities are smaller under the CAC with larger ratio. We note that the forced termination probability drops at $\lambda_1 = 13$ for ratio=10 and at $\lambda_1 = 10$ for ratio=100. The reason is that the number of calls existing in a cell, from which the optimal policy starts rejecting new calls for hand-off calls at those arrival rate, is decremented by one. The figure shows that the forced termination probabilities under the CAC with ratio=10 and ratio=100 are $4.5 \times 10^{-2}$ and $8.9 \times 10^{-3}$ times, respectively, larger than that under the non-CAC at $\lambda_1 = 15$.

We should not overlook the difference in the effects of CAC at low and high voice load conditions. Because the blocking probabilities whether under the CAC or under the non-CAC are very large at heavy voice load, the effect of CAC at heavy voice load is relatively small compared with that at light voice load in Figure 6. On the other hand, the effect of CAC on the forced termination probabilities at heavy voice load is larger than that at light voice load in Figure 7.

Figure 8 shows that the channel utilization under the CACs with ratio=10 and ratio=100 is 0.94 and 0.90 times, respectively, larger than that under the non-CAC at $\lambda_1 = 15$. This implies that channels for hand-off calls are underutilized under the CAC, while they are effectively utilized under the non-CAC.

Let us turn to look at the effect of CAC on the performance of data traffic. To begin with, our frame level model makes sure that the system ensure the target PEP for data whether the voice load is low or high. Figures 9–11 display the average packet throughput versus the packet arrival rate at low, middle and high voice load conditions, respectively. As the control of calls is not effective at low voice load condition, the throughput seems to be insensitive to the ratio. In the case of middle and high voice load, we observe that the CAC can boost the throughput significantly. Figure 10 shows that the CACs with ratio=10 and ratio=100 at middle voice load improve the throughput up to 1.9% and 8.4%, respectively, at $\lambda_d = 15$. In Figure 11, we find that the CACs with ratio=10 and ratio=100 at high voice load improve the throughput up to 19.6% and 32.4%, respectively, at $\lambda_d = 15$.

Suppose that the size of the data buffer is finite or that the average waiting time has to be upper bounded. Since the waiting time depends on the throughput, the effect of CAC on the waiting time is related with that on the throughput. Figures 12–14 show the average waiting time versus the packet arrival rate at low and high voice load conditions, respectively. Figure 12 shows that the waiting time seems to be insensitive to the ratio at low voice load. Figure 13 shows up to 7.1% and 25.5% improvement in the waiting time at $\lambda_d = 1$ under the CACs with ratio=10 and ratio=100, respectively, at middle voice load. In Figure 14, we find up to 34.4% and 49.2% improvement in the waiting time at $\lambda_d = 1$ under the CACs with ratio=10 and ratio=100, respectively, at high voice load. These results indicate us that if the voice load is low, then the waiting times under the CAC of different ratios are nearly equal; if the voice load is high, then the severer the CAC is the shorter the waiting time is.

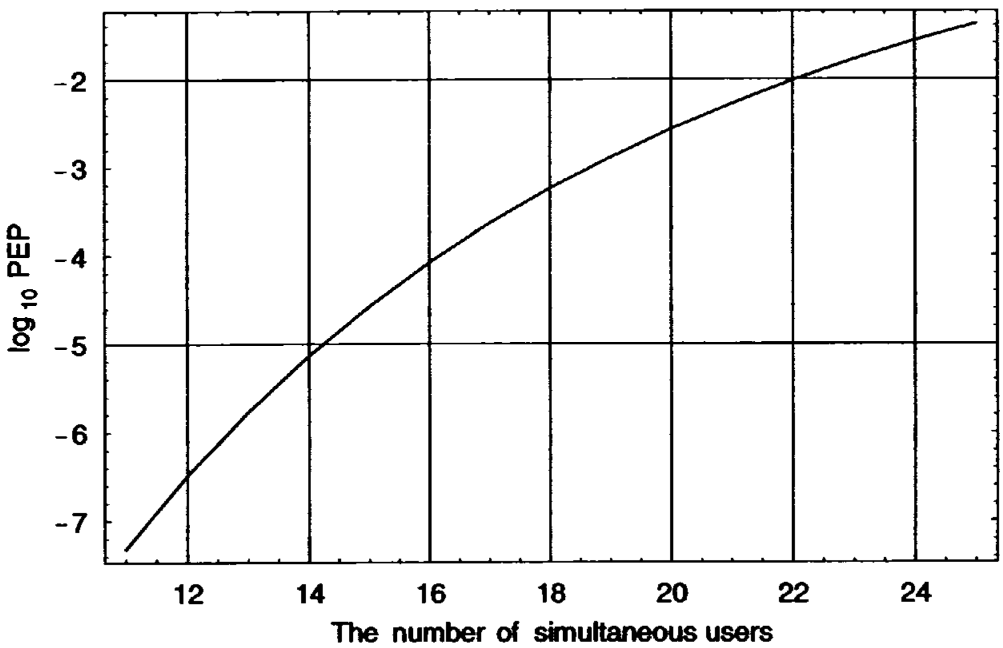


Figure 5: PEP versus MAC.

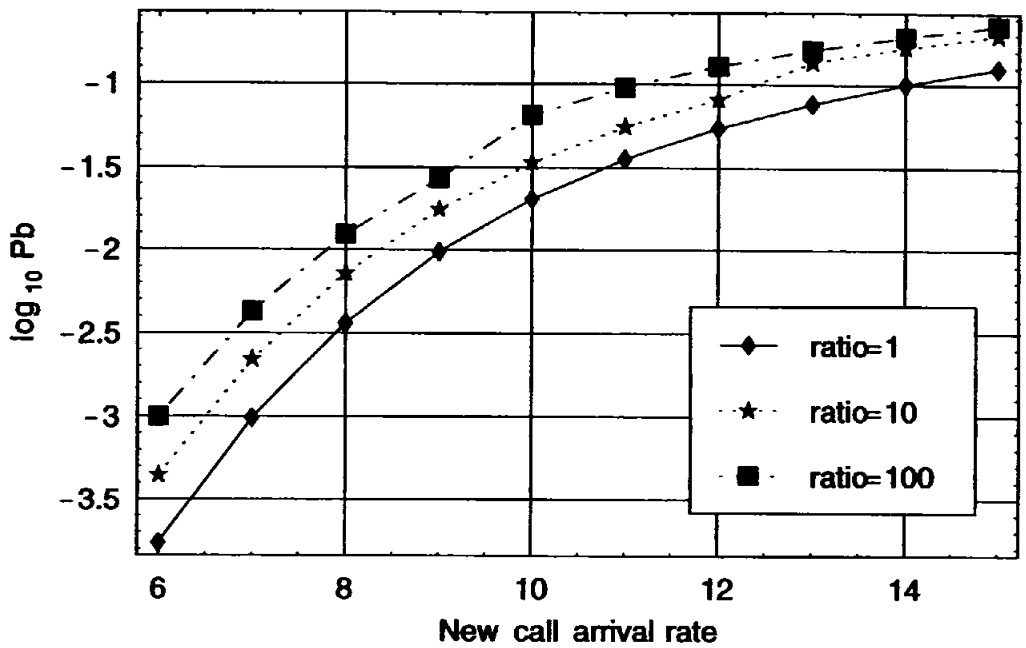


Figure 6: Blocking probability.

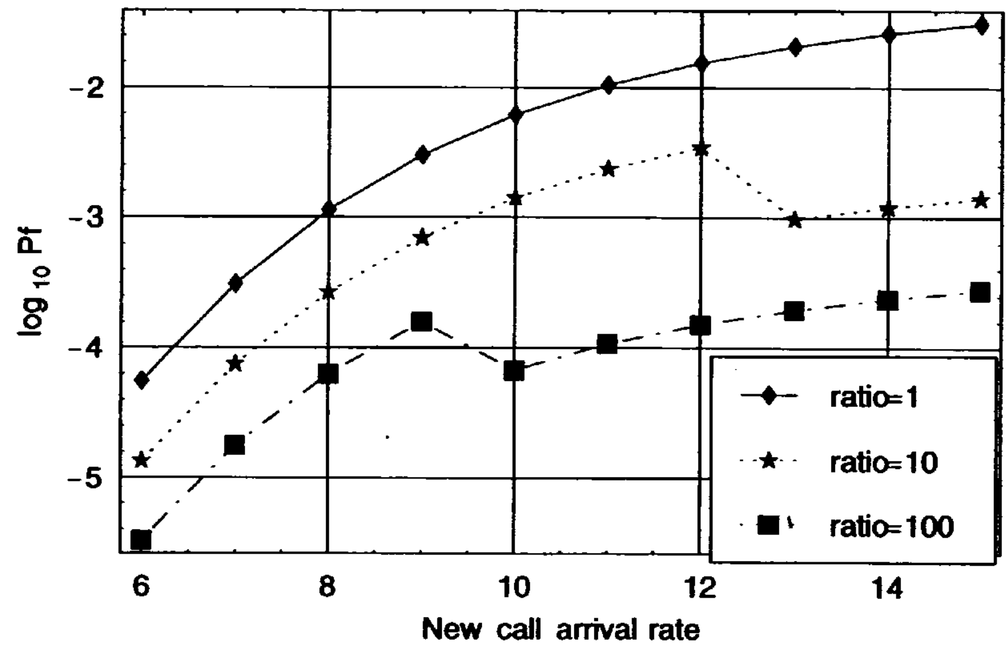


Figure 7: Forced termination probability.

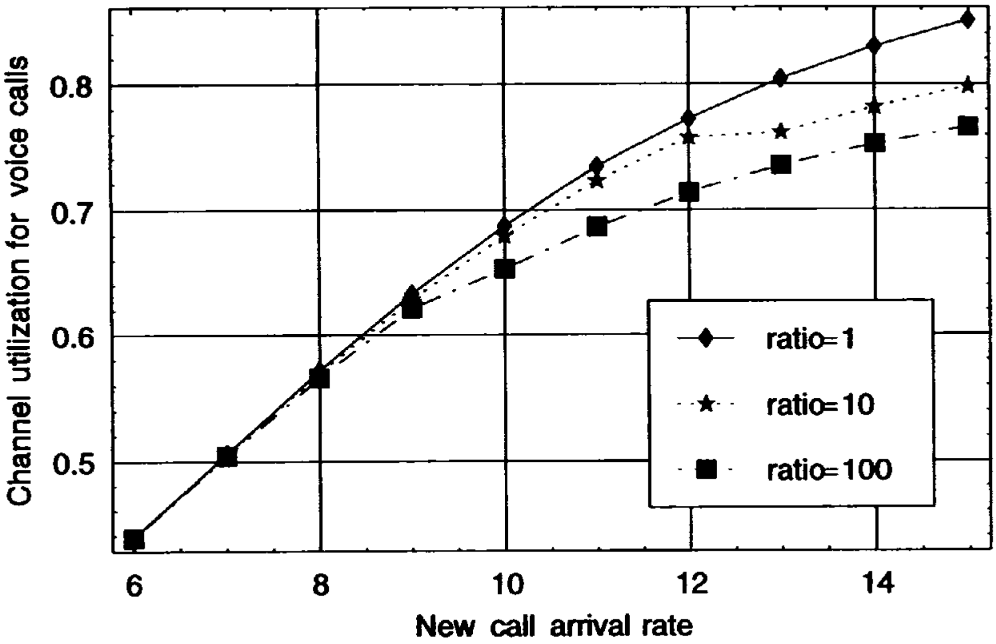


Figure 8: Channel utilization for voice calls.

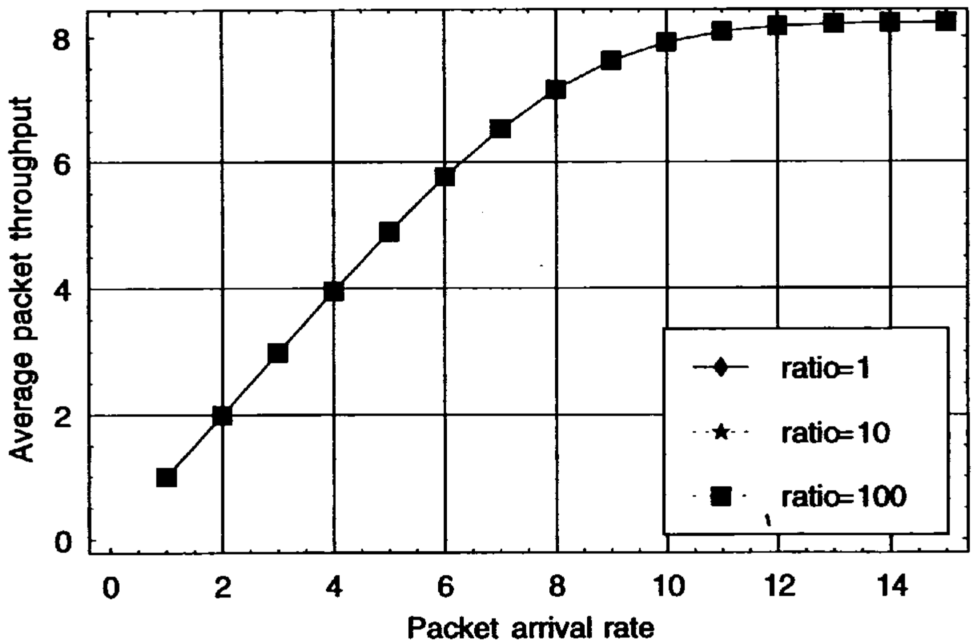


Figure 9: Packet throughput at low voice load ($\lambda_1 = 6$).

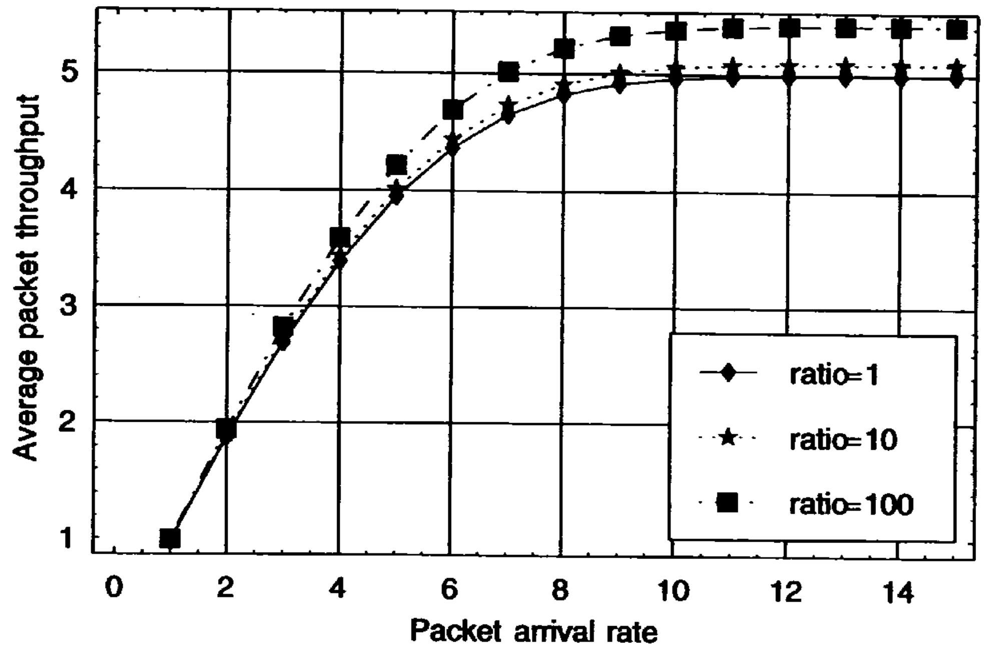


Figure 10: Packet throughput at middle voice load ($\lambda_1 = 10$).

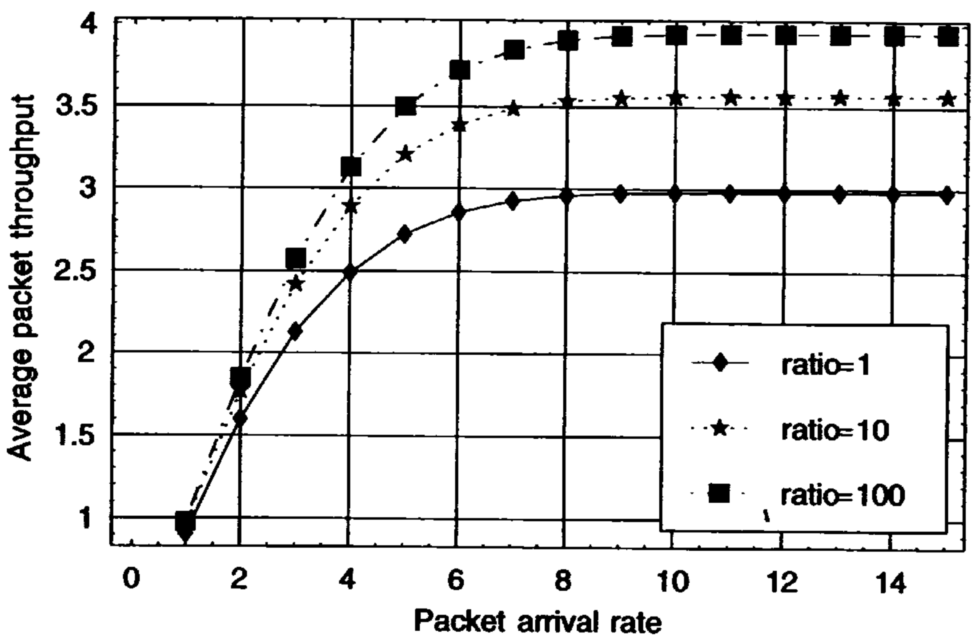


Figure 11: Packet throughput at high voice load ($\lambda_1 = 15$).

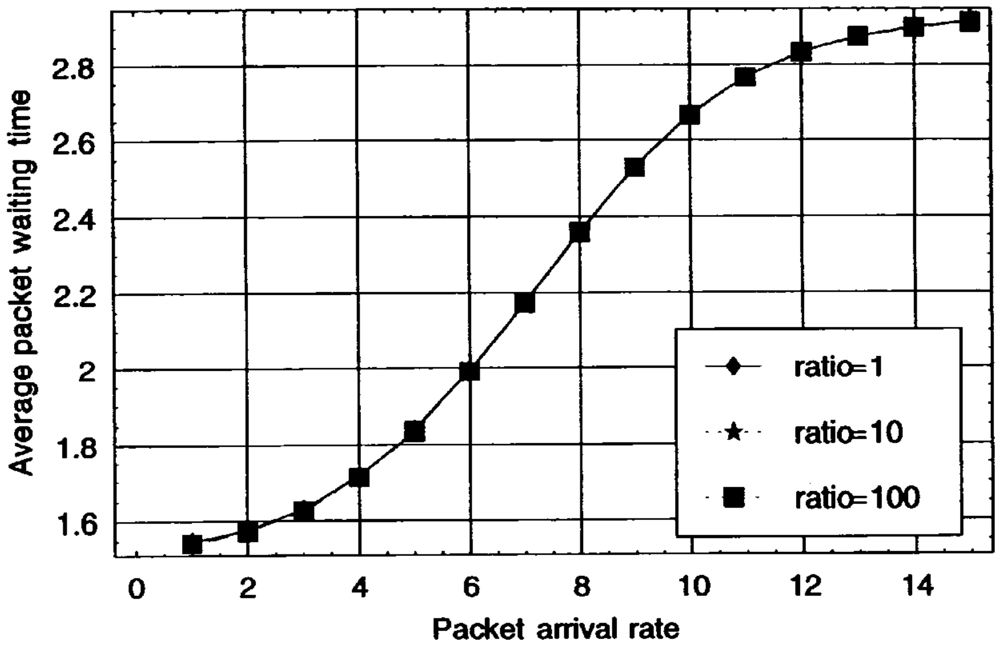


Figure 12: Packet waiting time at low voice load ($\lambda_1 = 6$).

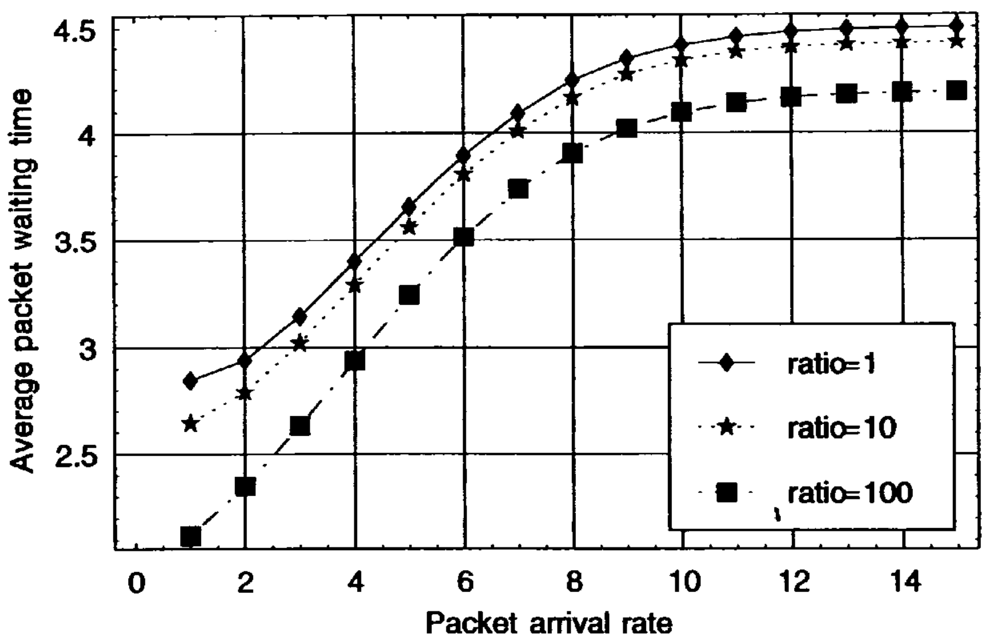


Figure 13: Packet waiting time at middle voice load ($\lambda_1 = 10$).

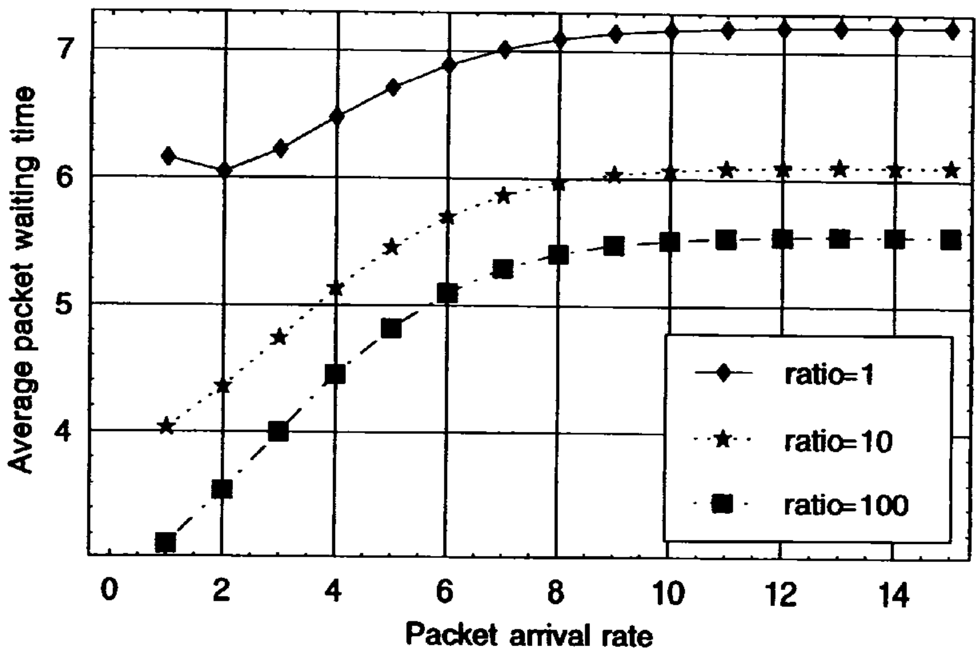


Figure 14: Packet waiting time at high voice load ($\lambda_1 = 15$).

# 5 Conclusions

In this paper we have proposed a CAC scheme for voice traffic on the uplink channel, and analyzed its effects on the integrated voice and data traffic in the downlink channel in mobile communication networks. Our CAC scheme in the call level brings good results to the forced termination probabilities for hand-off calls, while it worsens the blocking probabilities for new calls. In the frame level, the throughput and waiting time of data packets are improved at high voice load conditions.

Our analytic model has assumed that exactly one packet is served per frame for simplification. We need to tackle with the case which drops this assumption, because the transmission of one packet may practically go over multiple frames. Additionally, it would be interesting to consider the CAC scheme exercised on not only voice but also data traffic.

# References


[1] J. S. Lehnert and M. B. Pursley, "Error probabilities for binary direct-sequence spread spectrum communications with random signature sequences," IEEE Trans. Commun., vol. COM-35, no.1, pp.87–98, January 1987.

[2] R. K. Morrow, Jr. and J. S. Lehnert, "Bit-to-bit error dependence in slotted DS/SSMA packet systems with random signature sequences," IEEE Trans. Commun., vol.37, no.10, pp.1052–1061, October 1989.

[3] J. M. Holtzman, "A simple, accurate method to calculate spread-spectrum multiple-access error probabilities," IEEE Trans. Commun., vol.40, no.3, pp.461–464, March 1992.

[4] R. K. Morrow, Jr, "Accurate CDMA BER calculations with low computational complexity," IEEE Trans. Commun., vol.46, no.11, pp.1413–1417, November 1998.

[5] W. Yang and E. Geraniotis, "Admission policies for integrated voice and data traffic in CDMA packet radio networks," IEEE J. Select. Areas Commun., vol.12, no.4, pp.654–664, May 1994.

[6] T. Liu and J. A. Silvester, "Joint admission/congestion control for wireless CDMA systems supporting integrated services," IEEE J. Select. Areas Commun., vol.16, no.6, pp.845–857, August 1998.

[7] D. H. Hong and S. S. Rappaport, "Traffic model and performance analysis for cellular mobile radio telephone systems with prioritized and nonprioritized handoff procedures," IEEE Trans. Veh. Technol., vol.VT-35, no.3, pp.77–92, August 1986.

[8] K. Sakamaki and H. Takagi, "Call loss and forced termination probabilities in cellular radio communication networks," The Transaction of the Institute of Electronics, Information and Communication Engineers B-II, vol.J80-B-II, no.3, pp.231–238, March 1997 (in Japanese).

[9] H. Takagi, K. Sakamaki and T. Miyashiro, "Call loss and forced termination probabilities in cellular radio communication networks with irregular topologies," Proc. SPIE, vol.3530, pp.66–75, Boston, Massachusetts, Nov.2–4, 1998.

[10] M. Ohmikawa and H. Takagi, "Call loss Probabilities in CDMA cellular mobile communication networks," The Transaction of the Institute of Electronics, Information and Communication Engineers B, vol.J82-B, no.12, pp.2311–2319, December 1999 (in Japanese).

[11] H. Takagi, *Queueing Analysis: A Foundation of Performance Evaluation, Volume 1: Vacation and Priority Systems, Part 1.* The Netherlands: Elsevier Science Publishers B.V., 1991.

[12] H. C. Tijms, *Stochastic Modeling and Analysis: A Computational Approach.* Great Britain: John Wiley & Sons, 1986.

[13] J. Bather, "Optimal decision procedures for finite Markov chains. part II: communicating systems," Adv. Appl. Prob., 5, pp.521–540, 1973.

[14] P. J. Schweitzer and A. Federgruen, "Geometric convergence of value-iteration in multichain Markov decision problems," Adv. Appl. Prob., 11, pp.188–217, 1979.

[15] E. Lyghounis, I. Poretti and G. Monti, "Speech interpolation in digital transmission systems," IEEE Trans. Commun., vol. COM-22, no.9, pp.1179–1189, September 1974.

[16] T. Ojanperä, R. Prasad, ed., *Wideband CDMA for Third Generation Mobile Communications.* The United States of America: Artech House, 1998.